CLAIMS

1. A coding method to facilitate the reproduction as sound of digitized speech signals transmitted to a user in a telecommunications system during a VOIP telephone call between the user terminals via a packet transmission network, in particular the Internet, the speech signals picked up by a terminal being coded digitally in accordance with a coding protocol which divides them temporally into a succession of segments of the same duration before converting them segment by segment into the form of packets which are transmitted via the transmission network to a destination terminal in which the packets are decoded using a decoding protocol complementary to the coding protocol to enable reproduction of the speech signals from reproduced signal segments, eliminating any packets transmitted twice and using a dissimulation algorithm for signal segments corresponding to missing packets, wherein segments of a succession being coded for transmission in the form of packets are analyzed to determine whether any segment is critical, i.e. likely not to be replaced effectively by a dissimulation algorithm in the destination terminal if the corresponding packet is missing, and/or whether it is to be considered as replaceable by a dissimulation algorithm in the destination terminal under the same conditions.

2. A coding method according to claim 1 in which packets are duplicated for each critical segment in order to enable the sending terminal to transmit critical segments twice.

3. A coding method according to claim 1, wherein replaceable packets are suppressed in the sending terminal in a succession of packets relating to transmitted speech signal segments in order to control the packet transmission bandwidth.

4. A method according to claim 3, wherein the sending terminal maintains a constant transmit output bandwidth in the event of duplication of critical packets for double transmission by suppressing replaceable packets and substituting packets resulting from duplication for said replaceable packets prior to transmission.

5. A method according to claim 2, wherein any critical packet which corresponds to a signal segment having an estimated error value relative to at least the immediately preceding segment which is greater than an estimated error threshold value is duplicated and said error values are determined from predefined characteristics taken into account for the signal segments when they are coded.

6. A method according to claim 2, wherein an indication of the rate of loss of packets provided by the destination terminal is taken into account in the process of choosing packets to be duplicated in a sending terminal.

7. Telecommunications equipment, in particular a coder or a user terminal, provided with individual or common coding means adapted to be connected to a packet exchange network and to communicate via the network with compatible equipment by means of packets of digitized sound signals, in particular speech signals, produced in the context of a VOIP telephone call, said equipment having software and/or hardware means for digitally coding sound signals, in particular speech signals, that it must send in accordance with a particular protocol which temporally divides said signals into a succession of segments of the same duration after they are converted into the form of packets and before they are sent and for reproducing as sound segments of digitized sound signals which are sent to it in the form of packets, eliminating

any packets received twice and using a dissimulation algorithm for signal segments corresponding to any missing packets in a succession of received packets, the equipment including software means and hardware means for implementing the coding method according to claim 1.